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430 Rec'd PCT/PTO 29 DEC 1999

FORM PTO-1390 (REV. 5-93) U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371		CASE NO. 9683/58
		U.S. APPLICATION NO. (If known, see 37 C.F.R. 1.5) 09/462127
INTERNATIONAL APPLICATION NO. PCT/JP99/02802	INTERNATIONAL FILING DATE May 27, 1999	PRIORITY DATE CLAIMED May 27, 1998

TITLE OF INVENTION
SPEECH DECODER AND SPEECH DECODING METHOD

APPLICANT(S) FOR DO/EO/US
Nobuhiko Naka

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a FIRST submission of items concerning a filing under 35 U.S.C. 371
2. ☐ This is a SECOND or SUBSEQUENT submission of items concerning a filing under 35 U.S.C. 371
3. ☐ This express request to begin national examination procedures (35 U.S.C. 371(f)) at any time rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39(1).
4. ☐ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.
5. ☒ A copy of the International Application as filed (35 U.S.C. 371(c)(2)).
   a. ☐ is transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☒ has been transmitted by the International Bureau.
   c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).
6. ☒ A translation of the International Application into English (35 U.S.C. 371(c)(2)).
7. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3)).
   a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☐ have been transmitted by the International Bureau.
   c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
   d. ☒ have not been made and will not be made.
8. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
9. ☒ An oath or declaration of the inventor(s) (35 U.S.C. 371(c)(4)).
10. ☐ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)) and/or amendments under Article 34.

Items 11. to 16. Below concern other document(s) or information included:

11. ☐ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
12. ☒ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
13. ☐ A FIRST preliminary amendment.
    ☐ A SECOND or SUBSEQUENT preliminary amendment.
14. ☐ A substitute specification.
15. ☐ A change of power of attorney and/or address letter.
16. ☒ Other items or information: Power of Attorney

U.S. APPLICATION NO. (if known, see 37 C.F.R. 1.5) 09/462127	INTERNATIONAL APPLICATION NO. PCT/JP99/02802	CASE NO. 9683/58

17. ☒ The following fees are submitted:	CALCULATIONS	PTO USE ONLY
Basic National Fee (37 CFR 1.492(a)(1)-(5)): Search Report has been prepared by the EPO or JPO $840.00		
International preliminary **examination** fee paid to USPTO (37 CFR 1.492(2)(1)) $670.00		
No international preliminary examination fee paid to USPTO (37 CFR 1.482) but international **search** fee paid to USPTO (37 CFR 1.492(a)(2)) $760.00		
Neither international preliminary **examination** fee (37 CFR 1.482) nor international search fee (37 CFR 1.492(a)(3)) paid to USPTO $970.00		
International preliminary examination **fee** paid to USPTO (37 CFR 1.482) and all claims satisfied provisions of PCT Article 33(1)-(4) $96.00		
ENTER APPROPRIATE BASIC FEE AMOUNT	$840.00	
Surcharge of $130.00 for furnishing the oath or declaration later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492(e)).		

Claims	Number Filed	Number Extra	Rate		
Total Claims	8 -20 =		x $ 18.00		
Independent Claims	4 - 3 =	1	x $ 78.00	78.00	
Multiple dependent claim(s) if Applicable)		0	+ $260.00		
			TOTAL OF ABOVE CALCUATIONS =	$918.00	
Reduction by ½ for filing by small entity, if applicable. Verified Small Entity statement must also be filed. (Note 37 CFR 1.9, 1.27, 1.28)			SUBTOTAL =	$	
Surcharge of $130.00 for furnishing the English translation later than the ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492(f)).			TOTAL NATIONAL FEE=	$ $918.00	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31), $40.00 per property +			TOTAL FEES ENCLOSED=	40.00 $958.00	
				Amount to be refunded	$
				charged	$958.00

a. ☒ A check in the amount of $958.00 cover the above fees is enclosed.

b. ☐ Please charge my Deposit Account No. 23-1925 in the amount of $_____to cover the above fees. A duplicate copy of this sheet is enclosed.

c. ☒ The Assistant Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. 23-1925. A duplicate copy of this sheet is enclosed.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

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DESCRIPTION

SPEECH DECODER AND SPEECH DECODING METHOD

Technical Field

The present invention relates to a speech decoder and speech decoding method used in speech CODECs.

Background Art

Audio decoders which generate excited signals from coded speech signals input in units of frames and generate decoded speech signals from these excited signals are known. Of these types of speech decoders, in those which are adapted to low bit rate speech CODECs, the excited signals are treated with emphasis processing such as pitch emphasis processing or formant emphasis processing in order to improve the subjective sound quality of the decoded speech.

However, when frame errors occur in succession, the noise components are emphasized by these emphasis processes, thereby increasing the distortion and lowering the subjective sound quality.

Disclosure of the Invention

The present invention has been accomplished in view of the above considerations, and has the object of offering a speech decoder and speech decoding method capable of lightening the reduction of the subjective sound quality even when frame errors occur in succession.

In order to achieve this object, the present invention offers a speech decoder which generates excited signals from coded speech signals inputted in units of frames and generates decoded speech signals from these excited signals, characterized by comprising emphasis processing means for performing an emphasis process on said excited signals; error detecting

means for detecting frame errors in said coded speech signals; counting means for counting a number of times said frame errors occurred in succession and outputting the successive error frame number; and emphasis process prohibiting means for prohibiting said emphasis process due to said emphasis processing means when said successive error frame number exceeds a predetermined reference error frame number.

According to this speech decoder, an emphasis process is performed on the excited signals when the communication environment is good, and the successive error frame number is less than or equal to a predetermined reference error frame number. As a result, good decoded speech signals with high subjective sound quality are obtained. On the other hand, if the communication environment becomes bad and the successive error frame number exceeds the reference error frame number, the emphasis processing of the excited signals is prohibited. Therefore, distortions in the decoded speech signals which occur when emphasis processing is performed in such cases can be avoided before they occur.

Additionally, aside from prohibiting emphasis processing of excited signals when the successive error frame number has exceeded the reference error frame number, it is possible to control the amount of emphasis in the emphasis process in accordance with the successive error frame number.

Brief Description of the Drawings

Fig. 1 is a block diagram showing the structure of a speech decoder which is an embodiment of the present invention.

Fig. 2 is a block diagram showing a specific structure applying the same embodiment to a CS-ACELP type speech decoder.

Fig. 3 is a diagram for explaining a first modification example of this embodiment.

Fig. 4 is a diagram for explaining a second modification example of this embodiment.

Best Modes for Carrying Out the Invention

Next, a preferred embodiment of the present invention shall be described with reference to the drawings.

Fig. 1 is a block diagram showing the structure of a speech decoder 10 which is an embodiment of the present invention.

This speech decoder 10 comprises a decoding processing portion 11 and a emphasis process control portion 12.

Here, the decoding processing portion 11 is a device for decoding the received decoded speech signals (bitstream) BS and outputting the decoded speech signals SP.

This decoding processing portion 11 comprises an emphasis processing portion 15, a first switch SW1 and a second switch SW2.

The emphasis processing portion 15 performs emphasis processing with respect to the signals to be processed SPC based on the various parameters contained in the decoded speech signal, and outputs the resulting emphasized signals to be processed SEPC.

The first switch SW1 and second switch SW2 are switches for switching the signals to be processed SPC so as to be supplied to the latter-stage circuits through the emphasis processing portion 15, or so as to be supplied to the latter-stage circuits through the bypass BP.

Next, the emphasis process control portion 12 is a device for controlling whether or not to perform the emphasis processes in the decoding processing portion 11 based on frame error conditions of the coded speech signal BS.

This emphasis process control portion 12 comprises an error detecting portion 16 and a counter portion 17.

Here, the error detecting portion 16 is a device for detecting the frame errors of the coded speech signal BS and outputting error detection signals SER.

Additionally, the counter portion 17 counts the successive frame error number based on the error detection signals SER, and outputting an emphasis process control signal CE for switching the first switch SW1 and the second switch SW2 to the bypass BP side to prohibit emphasis processing when the successive frame error number exceeds a preset reference successive frame error number.

Next, the operations of the present embodiment will be described.

First, when the successive frame error number outputted from the counter portion 17 is less than or equal to a preset reference successive frame error number, the first switch SW1 and second switch SW2 are set to the emphasis process portion 15 side. Therefore, signals to be processed SPC generated from various parameters contained in the coded speech signal BS are supplied to the emphasis processing portion 15 of the decoding processing portion 11 via the first switch SW1 for emphasis processing. Then, the emphasized signals to be processed SEPC obtained by this emphasis process are outputted to the latter connected devices. As a result, a decoded speech signal SP with good subjective sound quality is obtained.

On the other hand, when the communication quality is degraded and the successive frame error number outputted from the counter portion 17 exceeds the reference successive frame error number, the first switch SW1 and second switch SW2 are set to the bypass BP side. As a result, the signals to be processed SPC generated by the parameters contained in the coded speech signal BS are outputted to latter-connected devices without being emphasis processed by the emphasis processing portion 15. Since the emphasis process is prohibited in this way when the successive frame error number is large, it is possible to reduce distortions generated by in the decoded speech signals SP.

Next, with reference to Fig. 2, a specific example of application of the present embodiment to a speech decoder in a CS-ACELP (Conjugate Structure Algebraic Code Excited Linear Prediction) type CODEC shall be explained. This type of CS-ACELP format speech coder and speech decoder are described, for example, in R. Salam et al., "Design and Description of CS-ACELP: A Toll Quality 8kb/s Speech Coder", IEEE Trans. on Speech and Audio Processing, vol. 6, no. 2, March 1998.

In Fig. 2, the speech decoder 20 comprises a parameter decoder 21. This parameter decoder 21 is a device decoding a pitch delay parameter group GP, a cobebook gain parameter group GG, a codebook index parameter group GC and an LSP (Line Spectrum Pair) index parameter group GL from the received coded speech signals (bitstream) BS.

Here, the codebook index parameter group GC includes a plurality of codebook index

parameters and a plurality of codebook code parameters.

Additionally, the speech decoder 20 comprises an adaptive code vector decoder 22, a fixed code vector decoder 23 and an adaptive preprocessing filter 25.

Here, the adaptive code vector decoder 22 is a device for outputting an adaptive code vector ACV corresponding to the pitch delay parameter group GP. More specifically, this adaptive code vector decoder 22 has a rewritable memory, and this memory contains a predetermined number of adaptive code vectors ACV which have been input in the past. The adaptive code vector decoder 22 takes the pitch delay parameter group GP as an index, reads an adaptive code vector ACV corresponding to this index from the memory, and outputs the result. Additionally, when the excited signal SEXC is reconstructed by the excited signal reconstruction portion 27 to be described later, this excited signal SEXC is written into the memory of the adaptive code vector decoder 22 as a new adaptive code vector ACV, and the oldest adaptive code vector ACV in the memory is eliminated.

The fixed code vector decoder 23 is a device for outputting an original fixed code vector FCV0 corresponding to the codebook index parameter group GC.

The adaptive code vector decoder 22 and the fixed code vector decoder 23 correspond to the codebook decoder 18 in Fig. 1.

The adaptive preprocessing filter 25 is a device which functions as an emphasizing process means for emphasizing the harmonic components of the decoded original fixed code vector FCV0, and outputs the result as a fixed code vector FCV.

Here, the first switch SW1 is provided in front of the adaptive preprocessing filter 25 in order to switch whether to supply the original fixed code vector FCV0 outputted from the fixed code vector decoder 23 to be supplied to the adaptive preprocessing filter 25 or to be supplied to the bypass BP. Additionally, the second switch SW2 is provided after the adaptive preprocessing filter 25 to select either the output terminal of the adaptive preprocessing filter 25 or the bypass BP for connection to the excited signal reconstruction portion 27. The first switch SW1 and second switch SW2 are switched by means of a preprocessing control signal CPR to be described later.

Furthermore, the speech decoder 20 comprises a gain decoder 24 and an LSP reconstruction portion 26.

The gain decoder 24 is a device for outputting an adaptive codebook gain ACG and a fixed codebook gain FCG based on a fixed code vector FCV (or original fixed code vector FCV0) and a codebook gain parameter group GG.

The LSP reconstruction portion 26 is a device for reconstructing the LSP coefficient CLSP based on the LSP index parameter group GL.

Further, the speech decoder 20 comprises an excited signal reconstruction portion 27, an LP synthesis filter 28, a postprocessing filter 29 and a bypass filter / upscaling portion 30.

Here, the excited signal reconstruction portion 27 is a device for reconstructing the excited signal SEXC based on adaptive code vector ACV, an adaptive codebook gain ACG, a fixed codebook gain FCG and a fixed code bector FCV (or original fixed code vector FCV0). This excited signal SEXC is written into the memory of the adaptive code vector decoder 22 as a new adaptive code vector ACV, and the oldest adaptive code vector ACV in the memory is eliminated.

The LP synthesis filter 28 is a device which performs an LP synthesis based on the excited signal SEXC and the LSP coefficient CLSP to reconstruct the speech signal SSPC.

The postprocessing filter 29 is a device for performing postprocess filtering of the speech signal SPC. This postprocessing filter 29 is constructed of three filters, a long-term postprocessing filter, a short-term postprocessing filter and a slope compensation filter. These three filters are serially connected in the order of long-term posprocessing filter to short-term postprocessing filter to slope compensation filter in the direction of input to output.

The bypass filter / upscaling portion 30 is a device for performing a bypass filtering process and an upscaling process with respect to the output signals of the postprocessing filter 29.

Additionally, the speech decoder 20 comprises an error detecting portion 31 and a counter portion 32.

Here, the error detecting portion 31 detects frame errors in the received coded speech

signals BS and outputs error detection signals SER.

Additionally, the counter portion 32 counts the successive frame error number based on the error detection signal SER, outputs a preprocessing control signal CPR for selecting the preprocessing filter 25 by means of the first switch SW1 and the second switch SW2 when the successive frame error number is less than or equal to a predetermined reference frame error number, and outputs a preprocessing control signal CPR for selecting the bypass BP by means of the first switch SW1 and the second switch SW2 when the successive frame error number has exceeded the predetermined reference frame error number.

Next, the operations of the speech decoder 20 shall be explained.

First, when the successive frame error number is less than or equal to the reference frame error number, the counter portion 32 switches the first switch SW1 and second switch SW2 to the adaptive preprocessing filter 25 by means of a preprocessing control signal CPR. As a result, the original fixed code vector FCV0 outputted from the fixed code vector decoder 23 is supplied to the adaptive preprocessing filter 25. Then, an emphasis process for emphasizing the harmonic components is performed on the original fixed code vector FCV0 in the adaptive preprocessing filter 25, and the resulting fixed code vector FCV is supplied to the gain decoder 24 and the excited signal reconstruction portion 27. Thus, a decoded speech signal SP with good subjective sound quality is obtained.

On the other hand, when the communication quality degrades and the successive frame error number outputted from the counter portion 32 exceeds the preset reference successive frame error number, the first switch SW1 and the second switch SW2 are set to the bypass BP side. As a result, the original fixed code vector FCV0 outputted from the fixed code vector decoder 23 is supplied to the gain decoder 24 and excited signal reconstruction portion 27 without undergoing an emphasis process by means of the adaptive preprocessing filter 25. Since the emphasis process is prohibited in this way when the successive frame error number is large, it is possible to reduce distortion which is generated in the decoded speech signal SP.

An embodiment of the present invention has been explained above, but various

examples of modifications to this embodiment can be considered.

Fig. 3 is a block diagram showing the structure of a speech decoder according to a first modification example. In Fig. 3, the parts which are the same as those in Fig. 1 are indicated by the same reference numerals.

In the above-described embodiment, emphasis processing is prohibited when the successive frame error number exceeds the predetermined reference successive frame error number. In contrast, in a speech decoder 30 according to a first modification example, the degree of the emphasis processing is controlled by controlling the filter gain of the preprocessing filter 25' for performing emphasis processing as shown in Fig. 3. That is, the counter portion 17' counts the successive frame error number, outputs a gain control signal SGC which makes the filter gain of the preprocessing filter 25' a normal value when this successive frame error number is less than or equal to a predetermined reference frame error number, and outputs a gain control signal SGC for making the filter gain of the preprocessing filter 25' less than usual when the successive frame error number exceeds the predetermined reference frame error number.

In this case as well, it is possible to reduce the distortions which are generated by performing emphasis processing when frame errors occur in succession, so as to enable the degradation of the subjective sound quality to be reduced.

Fig. 4 is a block diagram showing the structure of a speech decoder according to a second modification example. In Fig. 4, the parts which are the same as those in Fig. 1 are indicated by the same reference numerals.

In the speech decoder 40 of the second modification example, the deoding processing portion 41 is provided with a plurality of preprocessing filters 25'-1 to 25'-n, a first multiplexer MX1 and a second multiplexer MX2 as shown in Fig. 4.

Here, the amount of emphasis (e.g., corresponding to the filter gain) of the emphasis process performed by each of the preprocessing filters 25'-1 to 25'-n are different, the amount of emphasis in the preprocessing filter 25'-1 being the highest, and the amount of emphasis becoming lower in advancing to preprocessing filter 25'-2, preprocessing filter 25'-3 and so on.

Between the first multiplexer MX1 and the second multiplexer MX2, one route is selected from among these preprocessing filters 25'-1 to 25'-n and the bypass BP.

The counter portion 17" counts the number of successive frame errors, and supplies a selection signal SSEL for selecting the bypass BP or a preprocessing filter of an emphasis amount suited to the number of successive frame errors to the first multiplexer MX1 and the second multiplexer MX2.

In this second modification example, e.g. when the successive frame error number is "0", the preprocessing filter 25'-1 with the highest amount of emphasis is selected by the first multiplexer MX1 and second multiplexer MX2.

Then, if the communication environment worsens, preprocessing filters with lower amounts of emphasis are chosen such as preprocessing filter 25'-2 preprocessing filter 25'-3, . . . as the successive frame error number increases from "0" to "1", "2", . . .

In this way, the effects of switching of emphasis processing can be reduced because the amount of emphasis of the emphasis process can be switched in multiple steps in accordance with the successive frame error number.

In the above description, a case of a CS-ACELP type speech decoder was given as a specific example of the speech signal processing device. However, the present invention can be applied to speech signal processing devices of other formats such as speech decoders using APC (Adaptive Predictive Coding), APC-AB (APC with Adaptive Bit allocation), APC-MLQ, ATC (Adaptive Transform Coding), MPC (Multi Pulse Coding), LPC (Linear Prediction Coding), RELP (Residual Excited LPC) CELP (Code Excited LPC), LSP (Line Spectrum Pair Coding) or PARCOR as long as they are speech signal processing devices which perform emphasis processing.

CLAIMS

1. A speech decoder which generates excited signals from coded speech signals inputted in units of frames and generates decoded speech signals from the excited signals, said speech decoder comprising:

emphasis processing means for performing an emphasis process on said excited signals;

error detecting means for detecting frame errors in said coded speech signals;

counting means for counting a number of times said frame errors occurred in succession and outputting the successive error frame number; and

emphasis process prohibiting means for prohibiting said emphasis process due to said emphasis processing means when said successive error frame number exceeds a predetermined reference error frame number.

2. A speech decoder which generates excited signals from coded speech signals inputted in units of frames and generates decoded speech signals from these excited signals, said speech decoder comprising:

emphasis processing means for performing an emphasis process on said excited signals, capable of controlling the amount of emphasis of said emphasis process;

error detecting means for detecting frame errors in said coded speech signals;

counting means for counting a number of times said frame errors occurred in succession and outputting the successive error frame number; and

emphasis amount control means for controlling the amount of emphasis of said emphasis processing means in accordance with said successive error frame number.

3. A speech decoder according to claim 2, wherein:

said emphasis processing means comprises a plurality of emphasis processing portions with different emphasis amounts, and selecting means for selecting an emphasis processing

portion for performing the emphasis process on said excited signals from among said plurality of emphasis processing portions; and

said emphasis amount control means controls the selection of the emphasis processing portion by said selecting means in accordance with said successive error frame number.

4. A speech decoder according to claim 3, wherein

said emphasis processing means comprises a bypass for outputting coded speech signals absolutely without performing the emphasis processes of said plurality of emphasis processing portions;

said selecting means is capable of selecting said bypass as well as said plurality of emphasis processing portions; and

said emphasis amount control means controls said selecting means so as to output said coded speech signals through the bypass of said emphasis processing means when said successive error frame number exceeds a predetermined value.

5. A speech decoder according to claim 3, wherein:

said emphasis process selecting means controls the amount of emphasis of said emphasis processing means so as to reduce said emphasis amount when said successive frame error number is large.

6. A speech decoder according to claim 3, wherein:

said emphasis processing means is a filter for performing a filtering process on said excited signals; and

said emphasis amount control means controls the gain of the filtering process of said filter depending on said successive error frame number.

7. A speech decoding method for generating excited signals from coded speech signals inputted in units of frames and generating decoded speech signals from these excited signals,

the method comprising a process for counting a number of successive frames of received coded speech signals having coding errors; and prohibiting emphasis processing with respect to said coded speech signals when the number exceeds a predetermined reference error frame number.

8. A speech decoding method for generating excited signals from coded speech signals inputted in units of frames and generating decoded speech signals from these excited signals, the method comprising a process for counting a number of successive frames of received coded speech signals having coding errors; and controlling an amount of emphasis of the emphasis process on said coded speech signals in accordance with this number.

ABSTRACT

A decoding processing portion 11 of a speech decoder 10 is provided with an emphasis processing portion 15 for performing an emphasis process on signals to be processed (excited signals) SPC generated from coded speech signals BS. A counter portion 17 counts the number of times code errors occurred in successive frames of the coded speech signal BS, and outputs the successive frame error number. When the successive frame error number outputted form the counter portion 17 is less than or equal to a preset reference successive frame error number, a first switch SW1 and second switch SW2 are set to an emphasis processing portion 15 side. Accordingly, the signals to be processed SPC generated from various parameters included in the coded speech signals are supplied through the switch SW1 to the emphasis processing portion 15 of the decoding processing portion 11 to perform an emphasis process. Then, the emphasized signals to be processed SEPC obtained by this emphasis process are outputted through the switch SW2 to latter connected devices. As a result, decoded speech signals SP with good subjective sound quality are obtained. On the other hand, when the communication quality is degraded and the successive frame error number outputted from the counter portion 17 exceeds a preset reference successive frame error number, the first switch SW1 and second switch SW2 are set to a bypass BP side. Accordingly, the signals to be processed SPC generated from the various parameters contained in the coded speech signals are outputted to the latter connected devices without emphasis processing by the emphasis processing portion 15. In this way, emphasis processing is prohibited when the successive frame error number is large, thereby reducing distortion generated in the decoded speech signals SP.

FIG. 1

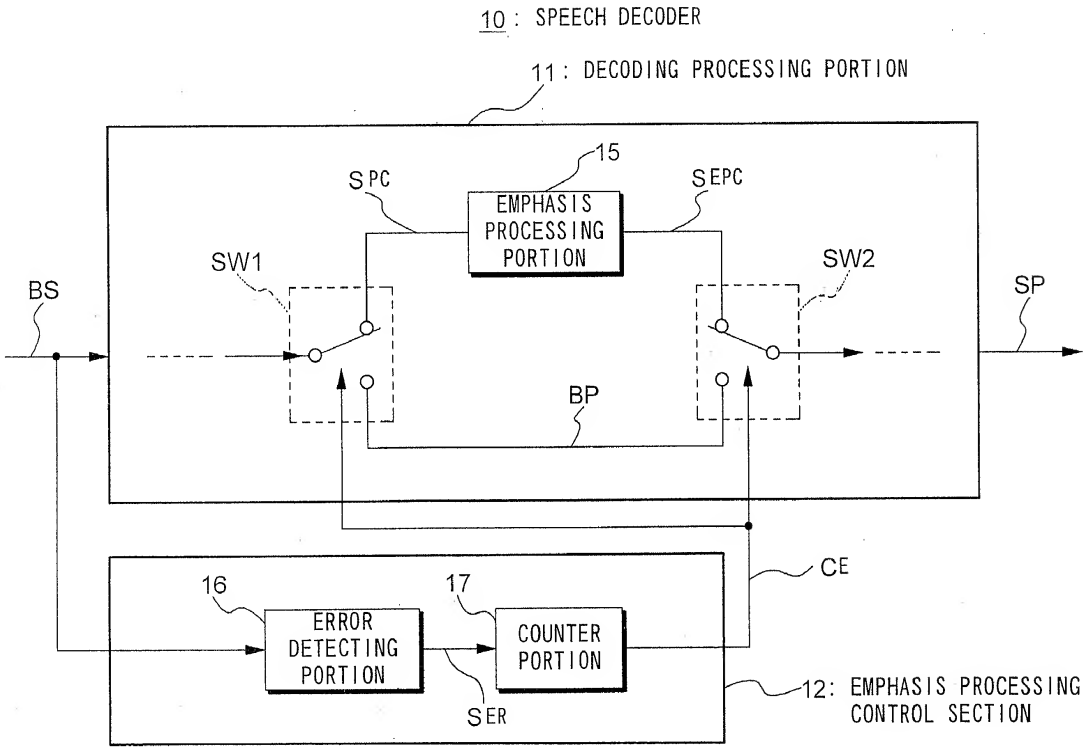


FIG. 2

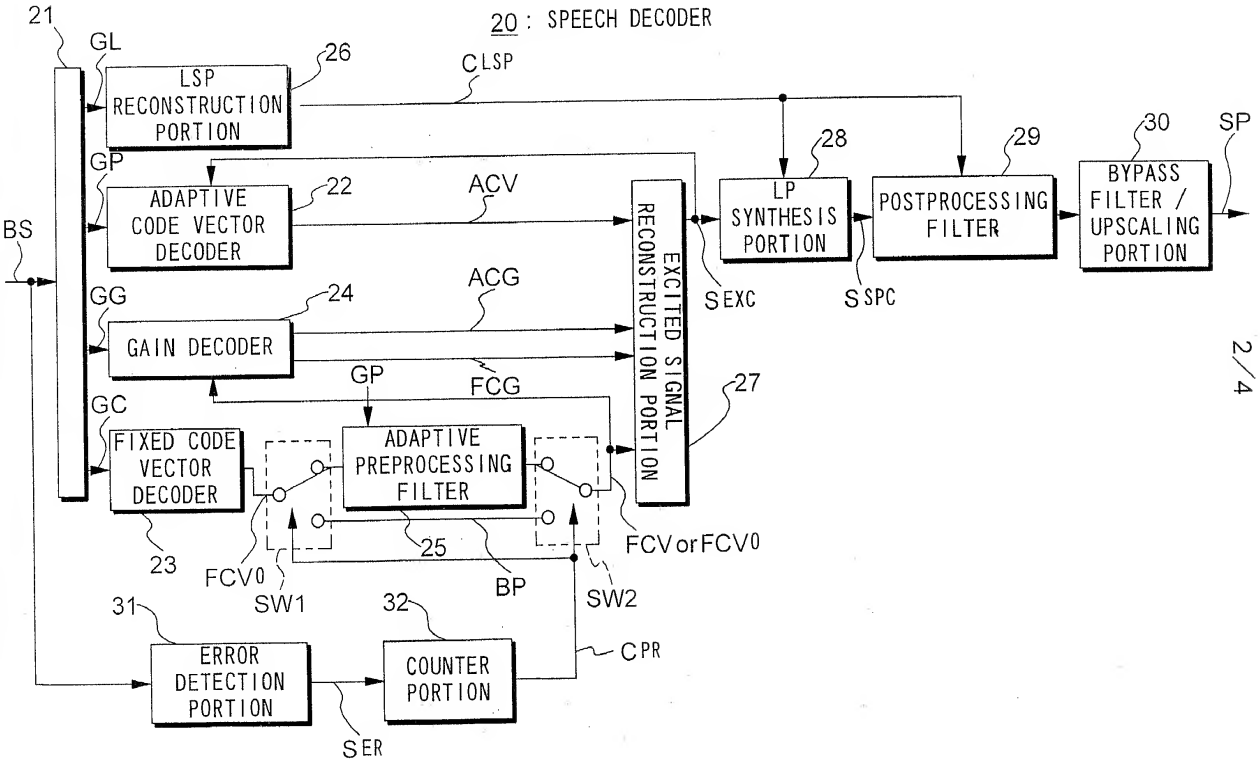


FIG. 3

30 : SPEECH DECODER

31 : DECODING PROCESSING PORTION

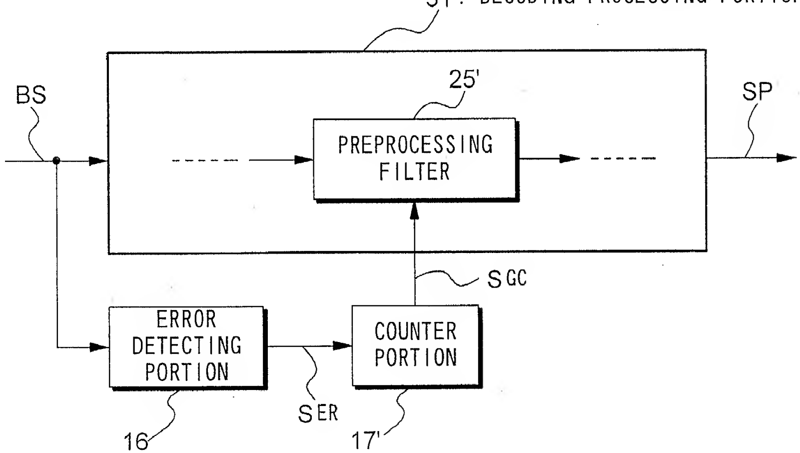
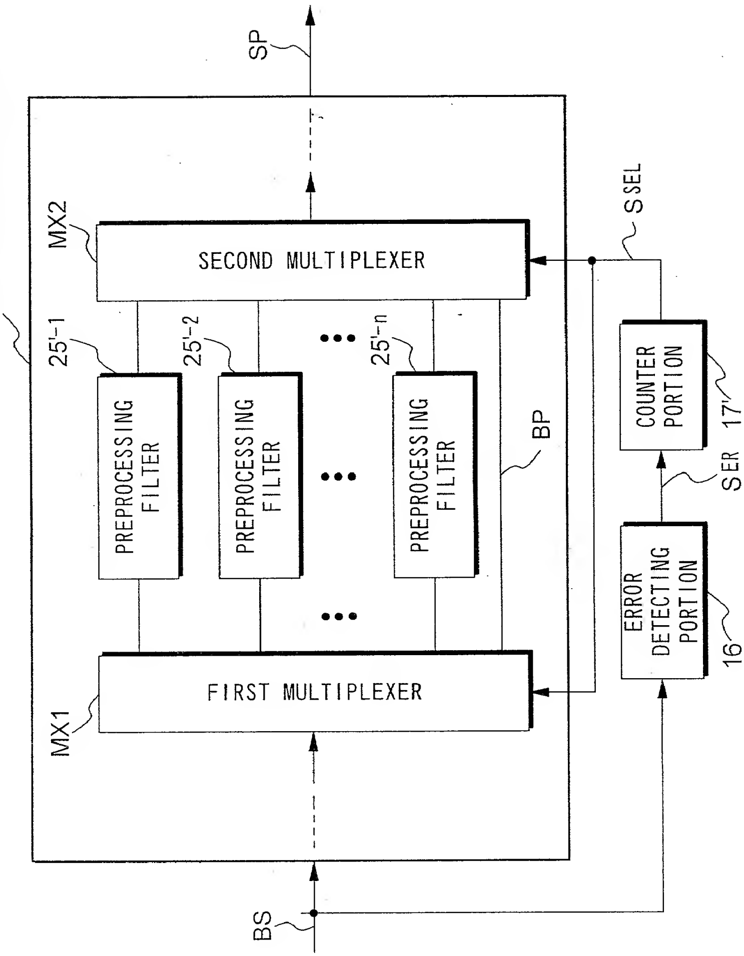


FIG. 4

40 : SPEECH DECODER

41 : DECODING PROCESSING PORTION



Case No. 9683/58

DECLARATION FOR PATENT APPLICATION

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name.

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled SPEECH DECODER AND SPEECH DECODING METHOD, the specification of which:

☐ is attached hereto.

☒ was filed on May 27, 1999 as International Application No. PCT/JP99/02802.

☐ and was amended on _____ (if applicable).

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the patentability as defined in Title 37, Code of Federal Regulations, § 1.56(a).

I hereby claim foreign priority benefits under 35 U.S.C. § 119(a)-(d) or § 365(b) of any foreign application(s) for patent or inventor's certificate or § 365(a) of any PCT International application which designated at least one country other than the United States, listed below and have also identified below, by checking the box, any foreign application for patent or inventor's certificate, or PCT International application having a filing date before that of the application on which priority is claimed:

Prior Foreign Application(s)			Priority Claimed	
Hei 10-146193	Japan	27/05/1998	☒	☐
(Number)	(Country)	(Day/Month/Year Filed)	Yes	No

I hereby claim the benefit under 35 U.S.C. § 119(e) of any United States provisional application(s) listed below:

(Application Serial No.)	(Filing Date)

I hereby claim the benefit under 35 U.S.C. § 120 of any United States application(s), or § 365(c) of any PCT International application designating the United States, listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States or PCT International application in the manner provided by the first paragraph of 35 U.S.C. § 112, I acknowledge the duty to disclose information which is material to patentability as defined in 37 CFR § 1.56 which became available between the filing date of the prior application and the national or PCT International filing date of this application:

PCT/JP99/02802	May 27, 1999	Pending
(Application Serial No.)	(Filing Date)	(Status-patented, pending, abandoned)

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Inventor's Signature Nobuhiko Naka Date: 13. Dec. 1999.

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Case No. 9683/58

Inventor(s): Nobuhiko Naka

Title: SPEECH DECODER AND SPEECH DECODING METHOD

POWER OF ATTORNEY

The specification of the above-identified patent application:

☐ is attached hereto
☒ was filed on May 27, 1999 as international application No. PCT/JP99/02802

I hereby revoke all previously granted powers of attorney in the above-identified patent application and appoint the following attorneys to prosecute said patent application and to transact all business in the Patent and Trademark Office connected therewith:

Gustavo Siller, Jr. - 32,305
Tadashi D. Horie - 40,437
Daniel B. Burg - 41,649

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The undersigned hereby authorizes the U.S. attorneys named herein to accept and follow instructions from _____ as to any action to be taken in the Patent and Trademark Office regarding this application without direct communication between the U.S. attorney and the undersigned. In the event of a change in the persons from whom instructions may be taken, the U.S. attorneys named herein will be so notified by the undersigned.

NTT Mobile Communications Network, Inc., a Japanese Corporation, certifies that it is the assignee of the entire right, title and interest in the patent application identified above by virtue of either:

☒ An assignment from the inventor(s) of the patent application identified above, a copy of which is attached hereto.
OR

☐ An assignment from the inventor(s) of the patent application identified above. The assignment was recorded in the Patent and Trademark Office at Reel _____, frame _____.
OR

☐ A chain of title from the inventor(s), of the patent application identified above, to the current assignee as shown below:

1. From _____ To:
The document was recorded in the Patent and Trademark Office at
Reel _____, frame _____, or a copy thereof is attached.

2. From _____ To:
The document was recorded in the Patent and Trademark Office at
Reel _____, frame _____, or a copy thereof is attached.

☐ Additional documents in the chain of title are listed on a supplemental sheet.

The undersigned has reviewed the assignment or all the documents in the chain of title of the patent application identified above and, to the best of undersigned's knowledge and belief, title is in the assignee identified above.

The undersigned (whose title is supplied below) is empowered to act on behalf of the assignee.

I hereby declare that all statements made herein of my own knowledge are true, and that all statements made on information and belief are believed to be true; and further, that these statements are made with the knowledge that willful false statements, and the like so made, are punishable by fine or imprisonment, or both, under Section 1001, Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

Signature 木下耕太 Date: 15/12/99

Name: Kota Kinoshita

Title: Senior Vice President,
Deputy Senior Executive Manager,
Research and Development Plan Department